

Listing of the Claims:

The following is a complete listing of all the claims in the application, with an indication of the status of each:

1 1 (Previously Amended). A speech coding apparatus including at least:

2 a spectrum parameter calculation section for receiving a speech

3 signal, obtaining a spectrum parameter, and quantizing the spectrum

4 parameter,

5 an adaptive codebook section for obtaining a delay and a gain from

6 a past quantized sound source signal by using an adaptive codebook, and

7 obtaining a residue by predicting a speech signal, and

8 a sound source quantization section for quantizing a sound source

9 signal of the speech signal by using the spectrum parameter and outputting

10 the sound source signal, comprising:

11 a discrimination section for discriminating a voiced sound mode

12 and an unvoiced sound mode on a basis of a past quantized gain of an

13 adaptive codebook;

14 a sound source quantization section which has a codebook for

15 representing a sound source signal by a combination of a plurality of non-

16 zero pulses and collectively quantizing amplitudes or polarities of the

17 pulses based on an output from said discrimination section, and searches

18 combinations of code vectors stored in said codebook and a plurality of

19 shift amounts used to shift positions of the pulses so as to output a

20 combination of a code vector and shift amount which minimizes distortion

21 relative to input speech; and

22 a multiplexer section for outputting a combination of an output

23 from said spectrum parameter calculation section, an output from said

24 adaptive codebook section, and an output from said sound source

25 quantization section.

1 2 (Previously Amended). A speech coding apparatus including at least:
2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter,
4 an adaptive codebook section for obtaining a delay and a gain from
5 a past quantized sound source signal by using an adaptive codebook, and
6 obtaining a residue by predicting a speech signal, and
7 a sound source quantization section for quantizing a sound source
8 signal of the speech signal by using the spectrum parameter and outputting
9 the sound source signal, comprising:
10 a discrimination section for discriminating a voice soundmode and
11 an unvoiced sound mode on a basis of a past quantized gain of an adaptive
12 codebook;
13 a sound source quantization section which has a codebook for
14 representing a sound source signal by a combination of a plurality of non-
15 zero pulses and collectively quantizing amplitudes or polarities of the
16 pulses based on an output from said discrimination section, and outputs a
17 code vector that minimizes distortion relative to input speech by generating
18 positions of the pulses according to a predetermined rule; and
19 a multiplexer section for outputting a combination of an output
20 from said spectrum parameter calculation section, an output from said
21 adaptive codebook section, and an output from said sound source
22 quantization section.

1 3 (Previously Amended). A speech coding apparatus including at least:
2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter, and quantizing the spectrum
4 parameter,
5 an adaptive codebook section for obtaining a delay and a gain from a
6 past quantized sound source signal by using an adaptive codebook, and
7 obtaining a residue by predicting a speech signal, and

8 a sound source quantization section for quantizing a sound source
9 signal of the speech signal by using the spectrum parameter and outputting
10 the sound source signal, comprising:

11 a discrimination section for discriminating a voice sound mode and
12 an unvoiced sound mode on the basis of a past quantized gain of an
13 adaptive codebook;

14 a sound source quantization section which has a codebook for
15 representing a sound source signal by a combination of a plurality of non-
16 zero pulses and collectively quantizing amplitudes or polarities of the
17 pulses based an output from said discrimination section, and a gain
18 codebook for quantizing gains, and searches combinations of code vectors
19 stored in said codebook, a plurality of shift amounts used to shift positions
20 of the pulses, and gain code vectors stored in said gain codebook so as to
21 output a combination of a code vector, shift amount, and gain code vector
22 which minimizes distortion relative to input speech; and

23 a multiplexer section for outputting a combination of an output
24 from said spectrum parameter calculation section, an output from said
25 adaptive codebook section, and an output from said sound source
26 quantization section.

1 4 (Previously Amended). A speech coding apparatus including at least:

2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter, and quantizing the spectrum
4 parameter,

5 an adaptive codebook section for obtaining a delay an a gain from a
6 past quantized sound source signal by using an adaptive codebook, and
7 obtaining a residue by predicting a speech signal, and

8 a sound source quantization section for quantizing a sound source
9 signal of the speech signal by using the spectrum parameter and outputting
10 the sound source signal, comprising:

11 a discrimination section for discriminating a voice sound mode and
12 an unvoiced sound mode on the basis of a past quantized gain of an
13 adaptive codebook;

14 a sound source quantization section which has a codebook for
15 representing a sound source signal by a combination of a plurality of non-
16 zero pulses and collectively quantizing amplitudes or polarities of the
17 pulses based on an output from said discrimination section indicates a
18 predetermined mode, and a gain codebook for quantizing gains, and
19 outputs a combination of a code vector and gain code vector which
20 minimizes distortion relative to input speech by generating positions of the
21 pulses according to a predetermined rule; and

22 a multiplexer section for outputting a combination of an output
23 from said spectrum parameter calculation section, an output from said
24 adaptive codebook section, and an output from said sound source
25 quantization section.

5 (Canceled).

1 6 (Previously Amended). A speech coding/decoding apparatus comprising:
2 a speech coding apparatus including:
3 a spectrum parameter calculation section for receiving a speech
4 signal, obtaining a spectrum parameter, and quantizing the spectrum
5 parameter,
6 an adaptive codebook section for obtaining a delay and a gain from
7 a past quantized sound source signal by using an adaptive codebook, and
8 obtaining a residue by predicting a speech signal,
9 a sound source quantization section for quantizing a sound source
10 signal of the speech signal by using the spectrum parameter and outputting
11 the sound source signal,
12 a discrimination section for discriminating a voice sound mode and

13 an unvoiced sound mode on the basis of a past quantized gain of a adaptive
14 codebook, and
15 a codebook for representing a sound source signal by a
16 combination of a plurality of non-zero pulses and collectively quantizing
17 amplitudes or polarities of the pulses when an output from said
18 discrimination section indicates a predetermined mode,
19 said sound source quantization section searching combinations of
20 code vectors stored in said codebook and a plurality of shift amounts used
21 to shift positions of the pulses so as to output a combination of a code
22 vector and shift amount which minimizes distortion relative to input
23 speech, and further including
24 a multiplexer section for outputting a combination of an output
25 from said spectrum parameter calculation section, an output from said
26 adaptive codeboook section, and an output from said sound source
27 quantization section; and
28 a speech decoding apparatus including at least:
29 a demultiplexer section for receiving and demultiplexing a
30 spectrum parameter, a delay of an adaptive codebook, a quantized gain,
31 and quantized sound source information,
32 a mode discrimination section for discriminating a mode by using a
33 past quantized gain in said adaptive codebook,
34 a sound source signal reconstructing section for reconstructing a
35 sound source signal by generating non-zero pulses from the quantized
36 sound source information when an output from said discrimination
37 indicates a predetermined mode, and
38 a synthesis filter section which is constituted by spectrum
39 parameters and reproduces a speech signal by filtering the sound source
40 signal.

1 7 (Previously Amended). A speech coding/decoding apparatus comprising:

2 a speech coding apparatus including:
3 a spectrum parameter calculation section for receiving a speech
4 signal, obtaining a spectrum parameter, and quantizing the spectrum
5 parameter,
6 an adaptive codebook section for obtaining a delay and a gain from
7 a past quantized sound source signal by using an adaptive codebook, and
8 obtaining a residue by predicting a speech signal,
9 a sound source quantization section for quantizing a sound source
10 signal of the speech signal by using the spectrum parameter and outputting
11 the sound source signal,
12 a discrimination section for discriminating a voice sound mode and
13 an unvoiced sound mode on the basis of a past quantized gain of an
14 adaptive codebook, and
15 a codebook for representing a sound source signal by a
16 combination of a plurality of non-zero pulses and collectively quantizing
17 amplitudes or polarities of the pulses based on an output from said
18 discrimination section,
19 said sound source quantization section outputting a combination of
20 a code vector and shift amount which minimizes distortion relative to input
21 speech by generating positions of the pulses according to a predetermined
22 rule, and further including
23 a multiplexer section for outputting a combination of an output
24 from said spectrum parameter calculation section, an output from said
25 adaptive codebook section, and an output from said sound source
26 quantization section; and
27 a speech decoding apparatus including at least:
28 a demultiplexer section for receiving and demultiplexing a
29 spectrum parameter, a delay of an adaptive codebook, a quantized gain,
30 and quantized sound source information,
31 a mode discrimination section for discriminating a mode by using a

32 past quantized gain in said adaptive codebook,
33 a sound source signal reconstructing section for reconstructing a
34 sound source signal by generating positions of pulses according to a
35 predetermined rule and generating amplitudes or polarities for the pulses
36 from a code vector when an output from said discrimination section
37 indicates a predetermined mode, and
38 a synthesis filter section which includes spectrum parameters and
39 reproduces a speech signal by filtering the sound source signal.

1 8 (Previously Amended). A speech coding apparatus comprising:
2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter, and quantizing the spectrum
4 parameter;
5 means for obtaining a delay and a gain from a past quantized sound
6 source signal by using an adaptive codebook, and obtaining a residue by
7 predicting a speech signal; and
8 mode discrimination means for receiving a past quantized adaptive
9 codebook gain and performing mode discrimination associated with a
10 voiced/unvoiced mode by comparing the gain with a predetermined
11 threshold, and
12 further comprising:
13 sound source quantization means for quantizing a sound source
14 signal of the speech signal by using the spectrum parameter and outputting
15 the signal, and searching combinations of code vectors stored in a
16 codebook for collectively quantizing amplitudes or polarities of a plurality
17 of pulses in a predetermined mode and a plurality of shift amounts used to
18 temporally shift a predetermined pulse position so as to select a
19 combination of an index of a code vector and a shift amount which
20 minimizes distortion relative to input speech;
21 gain quantization means for quantizing a gain by using a gain

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22 codebook; and

23 multiplex means for outputting a combination of outputs from said
24 spectrum parameter calculation means, said adaptive codebook means, said
25 sound source quantization means, and said gain quantization means.

1 9 (Original). An apparatus according to claim 8, wherein said sound source
2 quantization means uses a position generated according to a predetermined
3 rule as a pulse position when mode discrimination indicates a
4 predetermined mode.

1 10 (Original). An apparatus according to claim 9, wherein when mode
2 discrimination indicates a predetermined mode, a predetermined number of
3 pulse positions are generated by random number generating means and
4 output to said sound source quantization means.

1 11 (Original). An apparatus according to claim 8, wherein when mode
2 discrimination indicates a predetermined mode, said sound source
3 quantization means selects a plurality of combinations from combinations
4 of all code vectors in said codebook and shift amounts for pulse positions
5 in an order in which a predetermined distortion amount is minimized, and
6 outputs the combinations to said gain quantization means, and

7 said gain quantization means quantized a plurality of sets of
8 outputs from said sound source quantization means by using said gain
9 codebook, and selects a combination of a shift amount, sound source code
10 vector, and gain code vector which minimizes the predetermined distortion
11 amount.